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SPECIFICATION

TO ALL WHOM IT MAY CONCERN:

        BE IT KNOWN THAT I, Takaaki Suga, a citizen of
Japan residing at Kawasaki-shi, Kanagawa, Japan have
invented certain new and useful improvements in

 METHOD AND ROUTER CHANGING FRAGMENT SIZE OF DATA PACKETS

of which the following is a specification : -

TITLE OF THE INVENTION

METHOD AND ROUTER CHANGING FRAGMENT SIZE OF DATA PACKETS

5 BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention generally relates to a router and a method of changing a fragment size of data packets, and particularly relates to a router

10 connected to a network conveying data packets and audio packets and a method of changing a fragment size of data packets that are supplied to the network

2. Description of the Related Art

15 Fig.1 is an illustrative drawing for explaining a VoIP (voice over internet protocol) router.

As shown in Fig.1, a VoIP router 11 is provided between a WAN (wide area network), a LAN

20 (local area network, and a PBX (private branch exchange) 10. The VoIP router 11 converts data signals and audio signals into packets when the data signals are supplied from the LAN, and the audio signals are supplied from the PBX 10, and sends the

25 packets to the WAN. When receiving data packets and audio packets from the WAN, the VoIP router 11 converts these packets into data signals and audio signals, which are then supplied to the LAN and the PBX 10, respectively.

30 The VoIP router 11 establishes interface with the LAN, the WAN, and the PBX 10.

In the VoIP router 11, there is a need to avoid making an audio frame waiting until transmission of a packet to the WAN is finished

35 where the packet may be such a long packet as that of FTP (file transfer protocol) or HTTP (hypertext transport protocol). To this end, such a long

packet is divided, and audio packets are inserted
therebetween. This is called fragmentation. The
VoIP router checks an MTU (maximum transfer unit)
size of the IP (Internet protocol) layer. When the
5   router receives a packet having a size exceeding the
MTU size, the router notifies, via ICMP (Internet
control message protocol), the source of the packet
that the excess size of the packet creates error and
how large the MTU size is. An apparatus at the
10   packet source adjusts the packet size to the MTU
size, and transmits packets having a shorter size.
        In this configuration, when the VoIP
router receives a packet having a size exceeding the
MTU size, the VoIP router arranges for the source to
15   transmit shorter packets matching the MTU size.
Alternatively, the VoIP router may change the packet
to a shorter packet that conforms to the MTU size.
        Data that is transmitted via FTP or HTTP
forms a packet as large as 1000 bytes, for example.
20   Audio packets, on the other hand, have a size that
is as small as a two-digit figure in byte.
        Fig.2 is an illustrative drawing for
explaining transmission of data from a router.
        As shown in Fig.2, a long packet D may be
25   divided into shorter packets D1 through D5, which
are then transmitted to the WAN while audio packets
V1 through V4 having priority are inserted between
the shorter packets D1 through D5. Even in this
case, transmission of the audio packets may be
30   delayed if the MTU size is relatively large, thereby
degrading audio quality.
        For example, if the MTU size is so large
that the data packets D1 through D5 are
significantly larger than the audio packets V1
35   through V5, the audio packets V1 through V3 are
delayed by the data packet D5, and the audio packet
V1 is further delayed by the data packet D4.

In general, the shorter the fragment size of data, the higher the audio quality is. However, improvement in the audio quality is achieved at the expense of the throughput of data communication. Accordingly, if sufficient audio quality is being maintained, the fragment size of data may be lengthened to boost the throughput of data communication.

Conventionally, the MTU size is fixed, and does not change dynamically to cope with situational changes.

Accordingly, there is a need for a scheme that can automatically change a fragment size of a data packet so as to keep audio quality within a predetermined range.

SUMMARY OF THE INVENTION

It is a general object of the present invention to provide a router and a method of changing a fragment size that substantially obviate one or more of the problems caused by the limitations and disadvantages of the related art.

Features and advantages of the present invention will be set forth in the description which follows, and in part will become apparent from the description and the accompanying drawings, or may be learned by practice of the invention according to the teachings provided in the description. Objects as well as other features and advantages of the present invention will be realized and attained by a router and a method particularly pointed out in the specification in such full, clear, concise, and exact terms as to enable a person having ordinary skill in the art to practice the invention.

To achieve these and other advantages and in accordance with the purpose of the invention, as embodied and broadly described herein, the invention

provides a method of changing a fragment size of data packets in a router where a data packet is divided into the data packets having the fragment size, and are transmitted to a network along with
5 audio packets, including the steps of acquiring, in the router, a parameter indicative of whether proper audio quality is maintained through transmission of the audio packets, and changing the fragment size of the data packets in response to the acquired
10 parameter.

In the method described above, the parameter that indicates whether proper audio quality is maintained is acquired, and is consulted to change the fragment size of the data packets.
15 This makes it possible to improve data throughput while securing proper audio quality.

According to the present invention, the parameter is selected from a wait time of the audio packets, a delay time of the network, the number of
20 congestion notices, and the number of audio calls. The wait time is a time period for which the audio packets wait in the router before being transmitted to the network. The delay time of the network is a time period that passes from transmission of a hello
25 packet to reception of the hello packet returning from the network. The number of congestion notices indicates how many times a congestion notice is received from the network during a predetermined time period. The number of audio calls indicates
30 the number of audio calls simultaneously taking place in the router. Use of one of these parameters makes it possible to improve data throughput while securing proper audio quality.

35 **BRIEF DESCRIPTION OF THE DRAWINGS**

Fig.1 is an illustrative drawing for explaining a VoIP router;

Fig.2 is an illustrative drawing for explaining transmission of data from a router;

Fig.3 is an illustrative drawing showing a system to which the present invention is applied;

5 Fig.4 is a block diagram showing configurations of VoIP routers and a gatekeeper of Fig.3;

Figs.5A and 5B are tables showing data structures of a gatekeeper table and a routing table, 10 respectively;

Fig.6 is a block diagram of the VoIP router;

Fig.7 is a flowchart of a first method of adjusting a fragment size;

15 Fig.8 is an illustrative drawing for explaining how to determine a fragment size based on a wait-time deviation;

Fig.9 is a flowchart of a second method of adjusting a fragment size;

20 Fig.10 is a flowchart of a third method of adjusting a fragment size;

Fig.11 is a flowchart of a fourth method of adjusting a fragment size; and

Fig.12 is a flowchart of a fifth method of 25 adjusting a fragment size.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

In the following, embodiments of the present invention will be described with reference 30 to the accompanying drawings.

Fig.3 is an illustrative drawing showing a system to which the present invention is applied.

The system of Fig.3 is made up from points A through E that are connected to a WAN 100. The 35 WAN 100 is comprised of dedicated lines, frame-relay networks, ATM networks, and the like. The point A is comprised of a PBX 21A, VoIP router 22A, a server

23, and a gatekeeper 24. The points B through E
have an identical configuration, and include PBXs
21B through 21E, VoIP routers 22B through 22E, and
personal computers 25B through 25E, respectively.

5   The VoIP routers 22A through 22E are
connected to each other via the WAN 100. The point
A plays a key role in the system of Fig.3, and
attends to inter-computer communication (e.g.,
between the server 23 of the point A and the

10   personal computer 25B of the point B) as well as
inter-PBX audio communication (e.g., between the PBX
21A of the point A and the PBX 21B of the point B
via the VoIP router 22A of the point A). The WAN
conveys both the data packets and the audio packets.

15   Fig.4 is a block diagram showing
configurations of the VoIP routers and the
gatekeeper.

        The VoIP router 22 converts data signals
and audio signals into IP frames, and transmits the

20   IP frames. In Fig.4, any one of the VoIP routers
22A through 22C includes a control unit 30, a
routing table 31, a WAN-interface unit 32, a routing
unit 33, an audio-interface unit 34, and a LAN-
interface unit 35. The LAN-interface unit 35 is

25   connected to the server 23 or the personal computer
25B or 25C via a LAN. The audio-interface unit 34
is connected to the PBX 21A, 21B, or 21C.

        The control unit 30 of the VoIP router
attends to overall control of the VoIP router. In

30   detail, the control unit 30 arranges for the LAN-
interface unit 35 to attend to packet-
dividing/assembling operation, and arranges for the
routing unit 33 to attend to packet-priority-control
operation. Further, the control unit 30 updates the

35   routing table 31 as it becomes necessary through
communication with other VoIP routers, and conducts
communication with the gatekeeper 24.

The LAN-interface unit 35 establishes
interface with a LAN such as 10/100BASE. In detail,
the LAN-interface unit 35 divides a long packet, and
assembles divided packets under the control of the
5   control unit 30.

The audio-interface unit 34 establishes
interface with the PBX 21A, 21B, or 21C. In detail,
the audio-interface unit 34 digitizes audio signals
and signaling signals, and hands the digitized
10   signals to the routing unit 33. Further, the audio-
interface unit 34 detects signaling information
(e.g., call-transmission information, call-reception
information, phone-number information, and so on),
and informs the control unit 30.

15   The routing unit 33 delivers received
packets to the WAN-interface unit 32, the audio-
interface unit 34, and the LAN-interface unit 35
according to their destinations. Selection of an
interface unit is made by referring to a routing
20   table by using an address portion of a packet header.
Further, the routing unit 33 has a queue for packet
transmission, and adjusts a transmission order and
transmission timings under the control of the
control unit 30.

25   The routing table 31 is a table that
stores correspondences between IP addresses and VoIP
routers.

Figs.5A and 5B are tables showing data
structures of a gatekeeper table and a routing table,
30   respectively.

As shown in Fig.5B, the VoIP routing table
includes network addresses, costs, and relay routers.

In the routing table 31 of the VoIP router
22A at the point A, for example, the address
35   127.0.1.1 is listed together with cost "0" and no
relay router. This address 127.0.1.1 indicates the
address of the server 23 provided at the point A.

The cost of the server 23 indicates the number of intervening routers from the VoIP router 22A to the server 23, and, thus, is zero in this case. Since there is no need for relaying, no entry is given in

5   the field for the relay router.

Further, the address 127.0.3.1 at the point A is listed together with cost "1" and two relay routers having addresses 128.0.3.1 and 129.0.3.1.

10   The routing table 31 of the point B and the routing table 31 of the point C are structured in the same manner as the routing table 31 of the point A.

The gatekeeper 24 includes a control unit

15   40, a gatekeeper table 41, a LAN-interface unit 42, and an address-notifying unit 43.

The control unit 40 of the gatekeeper 24 attends to overall control of the gatekeeper. In details, the control unit 40 detects current audio

20   communication conditions, and updates the gatekeeper table.

The gatekeeper table 41 is a table in which phone numbers are stored with matching IP addresses. Communication flags are also stored for

25   the purpose of management and control of audio communication conditions.

As shown in Fig.5A, the gatekeeper table 41 includes prefix numbers, VoIP-router addresses, and communication flags.

30   As shown in Fig.5A, the VoIP-router address of a PBX at the point A having the prefix number 7000 is 127.0.2.1. The VoIP-router address of a PBX at the point B having the prefix number 7001 is 128.0.2.1. The VoIP-router address of a PBX

35   at the point C having the prefix number 7002 is 129.0.2.1.

The gatekeeper table 41 is used for

controlling the prefix numbers. On the other hand, extension numbers are controlled by the PBX. A communication flag that is 1 indicates an ongoing status of communication, and a communication flag that is 0 indicates no current communication.

The LAN interface unit 42 establishes interface with a LAN such as 10/100BASE.

The address-notifying unit 43 refers to the gatekeeper table 41, and responds to an inquiry of a phone number or an IP address when it is issued from a VoIP router.

Operation of the configuration of Fig.4 will be described below with reference to an example in which communication is simultaneously conducted between the point A and point B and between the point A and the point C.

[Telephone Communication between A and B]

A phone call is made from a phone connected to the PBX 21A at the point A to a phone connected to the PBX 21B at the point B. A procedure for establishing this communication will be described below.

1. When a call is made from the phone connected to the PBX 21A of the point A to the phone at 7001-xxxx that is connected to the PBX 21B of the point B, the PBX 21A at the point A ascertains from the prefix of the call that the call is not directed to itself but directed to an outside station. The PBX 21A sends signaling information to the VoIP router 22A.

2. The audio-interface unit 34 of the VoIP router 22A forwards the signaling information to the control unit 30, and digitizes it.

3. The control unit 30 sends an inquiry to the gatekeeper 24 to learn an IP address of the VoIP router corresponding to the prefix number 7001.

4. The address-notifying unit 43 of the

gatekeeper 24 refers to the gatekeeper table 41 to obtain the IP address 128.0.2.1 of the audio-interface unit 34 of the VoIP router 22B corresponding to the prefix number 7001, and sends

5   the obtained IP address to the VoIP router 22A as a reply to the inquiry. Further, the control unit 40 of the gatekeeper 24 detects a start of audio communication between the VoIP router 22A and the VoIP router 22B, and sets a communication flag in

10   the relevant table.

       5. The control unit 30 of the VoIP router 22A sends the received IP address to the routing unit 33 when the IP address is received from the gatekeeper 24. The routing unit 33 at the point A

15   consults the routing table 31, and finds an IP address 127.0.3.1 as an address to which the call is directed. Then, a packet directed to the VoIP router 22B is generated, and is send to the WAN-interface unit 32 of the point A.

20   6. The WAN-interface unit 32 at the point A transmits the packet to the WAN 100.

       7. The WAN-interface unit 32 at the point B receives the packet from the VoIP router 22A, and passes the packet to the routing unit 33.

25   8. The routing unit 33 at the point B refers to the routing table 31 at the point B, and ascertains that the packet is directed to the audio-interface unit 34 of the point B. The packet is then sent to the audio-interface unit 34 of the

30   point B.

       9. The audio-interface unit 34 at the point B disassembles the packet, and converts the signaling information into an analog signal, which is then sent to the PBX 21B.

35   10. The PBX 21B makes a relevant phone start ringing. When a user picks up the phone, signaling information to that effect is sent to the

caller at the point A via the VoIP router 22A and
the PBX 21A. The caller at the point A leans that
his/her call is connected.

11. Audio communication is also converted
5   into packets in the same manner as the signaling
information, and these packets are exchanged between
the VoIP routers.

12. When the user hangs up after finishing
the call, the control unit 30 of the VoIP router 22A
10 on the caller side notifies the gatekeeper 24 of the
end of the call.

13. The control unit 40 of the gatekeeper
24 resets the flag in the relevant table in response
to the notice from the VoIP router 22A.
15   This ends the communication between the
point A and the point B.

A phone call from the point A to the point
C is processed in much the same manner as described
above, with the VoIP router 22C taking a place of
20 the VoIP router 22B.

Concurrently with the audio communication,
data communication can be conducted between the
server 23 and the personal computer 25 of the point
B or between the server 23 and the personal computer
25 25 of the point C. In practice, audio communication
and data communication coexist as they are conducted.

The present invention improves efficiency
of data communication while keeping constant the
transmission intervals of audio packets for the
30 purpose of securing audio quality. In order to keep
constant the transmission intervals of audio packets,
a long packet for data communication is evenly
divided into packets of a predetermined length. The
shorter the length of the data packets, the better
35 the audio quality is. Improvement of audio quality
comes at the expense of throughput of data
communication.

In order to enhance efficiency of data communication while securing audio quality, therefore, the present invention adjusts a length that divides a long packet according to the

5   procedure as follows.

[First Method]

This method determines a fragment size of data packets based on a wait time of an audio packet in queue where the wait time is measured by the VoIP

10 router.

In Fig.4, the routing unit 33 of the VoIP router 22A creates a queue for each session. The routing unit 33 of the VoIP router 22A measures a wait time of an audio packet in queue, and notifies

15 the control unit 30 of the measured wait time.

The control unit 30 computes an average deviation from tens or hundreds of measurements, and adjusts a fragment size by following the procedure as shown in Fig.7.

20 Fig.7 is a flowchart of a method of adjusting a fragment size.

At a step S10, a check is made as to whether the deviation falls within a predetermined range.

25 Fig.8 is an illustrative drawing for explaining how to determine the fragment size based on the deviation.

When the deviation continues to exceed a certain threshold (B) for more than a predetermined

30 time period as shown in a time period T2 in Fig.8, the control unit 30 ascertains that the transmission intervals of audio packets fluctuates so much as to make it difficult to maintain audio quality. The control unit 30 instructs the routing unit 33 to

35 make the fragment size smaller than a default size. The routing unit 33 reduces the MTU size, thereby making smaller the packet size by a factor of 0.X.

This corresponds to a step S11.

When the deviation continues to stay within the predetermined range as shown in a time period T3 in Fig.8, the control unit 30 instructs
5 the routing unit 33 to return the fragment size to the default size. The routing unit 33 returns the MTU size to the default size. This corresponds to a step S12.

When the deviation continues to fall below
10 a certain threshold (A) for more than a predetermined time period as shown in a time period T4 in Fig.8, the control unit 30 ascertains that the transmission intervals of audio packets fluctuates so little as to warrant an increase of data
15 throughput. The control unit 30 instructs the routing unit 33 to make the fragment size larger than the default size. The routing unit 33 enlarges the MTU size, thereby making larger the packet size by a factor of 1.X. This corresponds to a step S13.
20 As a result, data packets are divided by the default MTU size during the time periods T1 and T3 shown in Fig.8, whereas data packets are divided by 0.X times the default MTU size during the time period T2, and are divided by 1.X times the default
25 MTU size during the time period T4.

In this manner, the present invention can insure desired audio quality during the time period T2, and can improve data throughput during the time period T4.
30 In the above description, a deviation is obtained from measurements of a wait time of audio packets in queue, and, then, is compared with some thresholds. Alternatively, a wait time rather than the deviation may be used and compared with
35 thresholds.

[Second Method]

This method determines a fragment size of

data packets based on a delay time of a network where the delay time is measured by the VoIP router using a hello packet.

The control unit 30 of the VoIP router 22A exchanges hello packets at constant intervals with other VoIP routers by using the routing protocol.

The control unit 30 measures a response time as a time period that passes from transmission of a hello packet to reception of the hello packet returning from another VoIP router, and adjusts a fragment size by following the procedure as shown in Fig.9.

Fig.9 is a flowchart of a method of adjusting a fragment size.

At a step S10, a check is made as to whether the response time falls within a predetermined range.

When the response time continues to exceed a certain threshold for more than a predetermined time period, the control unit 30 ascertains that a delay time of the network has increased to make it difficult to maintain audio quality. The control unit 30 instructs the routing unit 33 to make the fragment size smaller than a default size. The routing unit 33 reduces the MTU size, thereby making smaller the packet size. This corresponds to a step S11.

When the response time continues to stay within the predetermined range, the control unit 30 instructs the routing unit 33 to return the fragment size to the default size. The routing unit 33 returns the MTU size to the default size. This corresponds to a step S12.

When the response time continues to fall below a certain threshold for more than a predetermined time period, the control unit 30 ascertains that the delay time of the network has

decreased to warrant an increase of data throughput. The control unit 30 instructs the routing unit 33 to make the fragment size larger than the default size. The routing unit 33 enlarges the MTU size, thereby making larger the packet size. This corresponds to a step S13.

As a result, data packets are divided by the default MTU size when the delay time of the network stays within the predetermined range. On the other hand, data packets are divided by smaller than the default MTU size when the delay time of the network is long, and are divided by larger than the default MTU size when the delay time of the network is short.

In this manner, the present invention can improve data throughput while insuring desired audio quality.

In the above description, the response time of the network is obtained from measurements of a time period that passes from transmission of audio packets to reception of the audio packets, and, then, is compared with some thresholds. Alternatively, a deviation of the response time may be obtained and compared with thresholds.

[Third Method]

This method determines a fragment size of data packets based on how many times a notice of network congestion is received.

In networks such as frame-relay networks, ATM networks, etc., when congestion occurs, the VoIP router 22A is notified of the congestion. As the WAN-interface unit 32 of the VoIP router 22A receives the notice of congestion, the WAN-interface unit 32 passes the notice to the control unit 30.

In response, the control unit 30 of the VoIP router 22A counts how many times the notice of congestion is received during a predetermined time

period, and adjusts a fragment size by following the
procedure as shown in Fig.10.

Fig.10 is a flowchart of a method of
adjusting a fragment size.

At a step S10, a check is made as to
whether the number of received congestion notices
falls within a predetermined range.

When the number of received congestion
notices continues to exceed a certain threshold for
more than a predetermined time period, the control
unit 30 ascertains that the network congestion has
worsened to such an extent as to make it difficult
to maintain audio quality. The control unit 30
instructs the routing unit 33 to make the fragment
size smaller than a default size. The routing unit
33 reduces the MTU size, thereby making smaller the
packet size. This corresponds to a step S11.

When the number of received congestion
notices continues to stay within the predetermined
range, the control unit 30 instructs the routing
unit 33 to return the fragment size to the default
size. The routing unit 33 returns the MTU size to
the default size. This corresponds to a step S12.

When the number of received congestion
notices continues to fall below a certain threshold
for more than a predetermined time period, the
control unit 30 ascertains that the network
congestion is so little as to warrant an increase of
data throughput. The control unit 30 instructs the
routing unit 33 to make the fragment size larger
than the default size. The routing unit 33 enlarges
the MTU size, thereby making larger the packet size.
This corresponds to a step S13.

As a result, data packets are divided by
the default MTU size when the number of congestion
notices stays within the predetermined range. On
the other hand, data packets are divided by smaller

than the default MTU size when the number of congestion notices is large, and are divided by larger than the default MTU size when the number of congestion notices is small.

5   In this manner, the present invention can improve data throughput while insuring desired audio quality.

In the above description, the number of received congestion notices is obtained by counting

10 how many times the notice of congestion is received from the network, and, then, is compared with some thresholds. Alternatively, a deviation of the number of congestion notices may be obtained and compared with thresholds.

15 [Fourth Method]

This method determines a fragment size of data packets based on the number of audio calls that is reported from an apparatus that counts such a number.

20   The gatekeeper 24 can check the number of audio calls taking place at each VoIP router by referring to the communication flags provided in the gatekeeper table 41. When the number of audio calls changes, the gatekeeper 24 notifies the control unit

25 30 of the number of audio calls.

In response, the control unit 30 of the VoIP router 22A adjusts a fragment size based on the number of audio calls as shown in Fig.11.

Fig.11 is a flowchart of a method of

30 adjusting a fragment size.

At a step S10, a check is made as to whether the number of calls falls within a predetermined range.

When the number of calls continues to

35 exceed a certain threshold for more than a predetermined time period, the control unit 30 ascertains that the number of audio packets has

increased to such a level as to make it difficult to maintain audio quality. The control unit 30 instructs the routing unit 33 to make the fragment size smaller than a default size. The routing unit
5   33 reduces the MTU size, thereby making smaller the packet size. This corresponds to a step S11.

When the number of calls continues stay within the predetermined range, the control unit 30 instructs the routing unit 33 to return the fragment
10   size to the default size. The routing unit 33 returns the MTU size to the default size. This corresponds to a step S12.

When the number of calls continues to fall below a certain threshold for more than a
15   predetermined time period, the control unit 30 ascertains that it is warranted to increase data throughput. The control unit 30 instructs the routing unit 33 to make the fragment size larger than the default size. The routing unit 33 enlarges
20   the MTU size, thereby making larger the packet size. This corresponds to a step S13.

As a result, data packets are divided by the default MTU size when the number of calls stays within the predetermined range. On the other hand,
25   data packets are divided by smaller than the default MTU size when the number of calls is large, and are divided by larger than the default MTU size when the number of calls is small.

As the number of audio calls that are
30   simultaneously taking place increases, the number of audio packets increases. This makes it necessary to divide data packets into smaller fragments in order to maintain a desired audio quality. The fourth embodiment of the present invention changes the
35   fragment size of data packets in response to the number of audio calls, thereby making it possible to improve data throughput while insuring desired audio

quality.

[Fifth Method]

This method determines a fragment size of data packets based on the number of audio calls that is counted by the VoIP router.

The control unit 30 of the VoIP router 22A can check the number of audio calls from the signaling information. The VoIP router 22A lets the control unit 30 count the number of audio calls. The control unit 30 of the VoIP router 22A adjusts a fragment size based on the number of audio calls as shown in Fig.12.

Fig.12 is a flowchart of a method of adjusting a fragment size.

At a step S10, a check is made as to whether the number of calls falls within a predetermined range.

When the number of calls continues to exceed a certain threshold for more than a predetermined time period, the control unit 30 ascertains that the number of audio packets has increased to such a level as to make it difficult to maintain audio quality. The control unit 30 instructs the routing unit 33 to make the fragment size smaller than a default size. The routing unit 33 reduces the MTU size, thereby making smaller the packet size. This corresponds to a step S11.

When the number of calls continues stay within the predetermined range, the control unit 30 instructs the routing unit 33 to return the fragment size to the default size. The routing unit 33 returns the MTU size to the default size. This corresponds to a step S12.

When the number of calls continues to fall below a certain threshold for more than a predetermined time period, the control unit 30 ascertains that it is warranted to increase data

throughput. The control unit 30 instructs the routing unit 33 to make the fragment size larger than the default size. The routing unit 33 enlarges the MTU size, thereby making larger the packet size.

5 This corresponds to a step S13.

As a result, data packets are divided by the default MTU size when the number of calls stays within the predetermined range. On the other hand, data packets are divided by smaller than the default

10 MTU size when the number of calls is large, and are divided by larger than the default MTU size when the number of calls is small.

According to the fifth embodiment, the present invention changes the fragment size of data

15 packets in response to the number of audio calls, thereby making it possible to improve data throughput while insuring desired audio quality.

[Detailed Operation]

In the following, details of operation of

20 the VoIP router will be described with reference to the first method.

Fig.6 is a block diagram of the VoIP router.

As previously described, the VoIP router

25 includes the control unit 30, the WAN-interface unit 32, the routing unit 33, the audio-interface unit 34, and the LAN-interface unit 35.

The routing unit 33 in Fig.6 includes a queue-wait-time-monitoring timer 50, a packet-

30 transmission unit 51, a queue 52, an IP-packet unit 53, and a fragmentation unit 54.

The queue-wait-time-monitoring timer 50 measures a wait time of an audio packet in queue, and sends the measurement to the control unit 30.

35 The packet-transmission unit 51 transmits audio packets ahead of other packets under the control of the control unit 30. The queue 52 has data packets

and audio packets waiting therein, and is provided
for each session under the control of the control
unit 30. The IP-packet unit 53 converts audio
signals into packets as the audio-interface unit 34
5 digitizes the audio signals. The fragmentation unit
54 divides data packets into fragments of a
predetermined size under the control of the control
unit 30.

LAN-data packets are received by the LAN-
10 interface unit 35 of the VoIP router 22A, and are
forwarded to the fragmentation unit 54 of the
routing unit 33. The fragmentation unit 54 breaks
the packets into fragments of proper sizes, which
are then sent to the queue 52. There are a
15 plurality of queues 52, each of which is prioritized.
In Fig.6, for example, higher priority is given to
the queues as the queues come closer to the bottom.
In the order of priority, the packet-transmission
unit 51 takes out packets from the queues 52, and
20 the WAN-interface unit 32 transmits these queues.

Packets each wait in the queues 52 until
their turn comes. A time period during which a
packet stays waiting in the queue is referred to as
a wait time in queue. When audio is transmitted as
25 packets, it is necessary to keep packet intervals
constant in order to maintain audio quality. It is
desirable, therefore, that a wait time in queue is
as short and constant as possible. A need for a
shorter wait time is satisfied by putting audio
30 packets in the queue that is given priority. As for
constancy, fluctuation of a waiting time in queue is
determined by how often data having a packet length
longer than audio packets are inserted between audio
packets during transmission.

35 When the wait time in queue fluctuates
violently, there is a need to shorten a fragment
size of data packets. When the wait time in queue

stays constant, on the other hand, it is reasonable to ascertain that audio quality is properly maintained, so that the fragment size is increased with an aim of improving data throughput.

5   Further, the present invention is not limited to these embodiments, but various variations and modifications may be made without departing from the scope of the present invention.

The present application is based on
10 Japanese priority application No. 11-229468 filed on August 13, 1999, with the Japanese Patent Office, the entire contents of which are hereby incorporated by reference.